Project Proposal for Software Defined Radio

Advisors: Dr. Stewart & Dr. Ahn

Luke Vercimak
Karl Weyeneth

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Bradley University ECE Department

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Summary

A software defined radio is a radio transmitter/receiver that uses digital signal processing (DSP) for coding/decoding and modulation/demodulation. Using digital signal processing for the radio allows for greater flexibility and accuracy when designing radios. This project will focus on the design and implementation of a digital software radio. The project will be a scaled down version of the 802.11a standard and will use QAM (Quadrature Amplitude Modulation) and will be multiplexed with OFDM (Orthogonal Frequency Division Multiplexing). The project will also focus on rapid development and prototyping by using Simulink block diagrams to program the Texas Instruments TMDSK6713 evaluation board.

Functional Description

A software defined radio is a radio transmitter/receiver that uses digital signal processing (DSP) for coding/decoding and modulation/demodulation. This allows much more power and flexibility when choosing and designing modulation and coding techniques. The C6700 series of digital signal processors have been chosen for this project. More specifically the TMDSK6713 evaluation board with the TMS320C6713 DSP chip will be used.

Due to hardware availability, both the transmitter and receiver will be implemented on the same DSP evaluation board. The system will be constructed and programmed entirely in Simulink using the embedded target for TI C6000 Simulink library. Simulink will generate the code based off of the model designed and will then download it to the board through TI Code Composer Studio for testing.

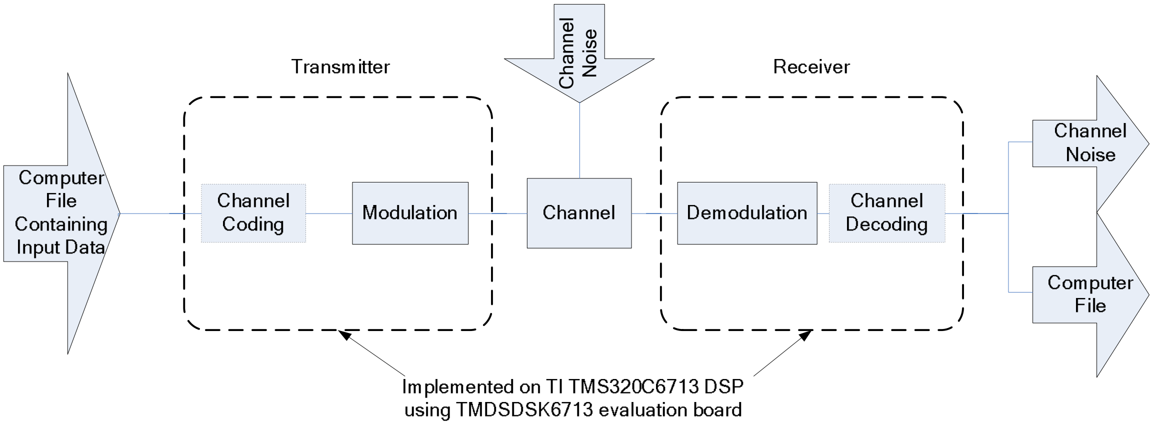

Figure 1 - I/O block diagram for transmitter and receiver radio systems

Inputs and Outputs

An overall block diagram for the software radio project is found in figure 1. The inputs to the system are a digital data source (computer file) and channel noise. The output of the system is the recovered input data. The recovered data should be received exactly as transmitted. This can be displayed on an oscilloscope coming out of the DSP evaluation board and/or stored on a computer file for further verification and analysis.

Modes of Operation

The input from the digital data source will be sent into the transmitter. There it will have channel coding applied to provide protection from data corruption introduced by noise. This part will not be implemented in this project. After that, the encoded digital signal will then be modulated with an appropriate modulation technique and transmitted through the channel. An appropriate model and representation for the channel also needs determined. After this, the receiver demodulates the signal and applies appropriate channel decoding. From there the reconstructed digital signal will be available for further analysis.

System Block Diagram

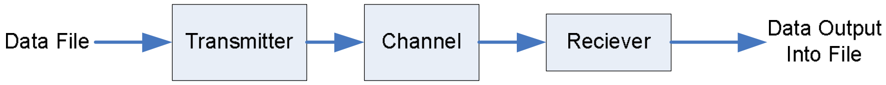

Figure 2 - System Breakdown of the Software Radio

The input to the system will be digital data in a computer file. This data will be modulated by the transmitter and sent to the channel. The channel will cause interference to the signal in the form of attenuation, phase delay, and noise. At the receiver side, the signal will be demodulated and reconstructed to produce the original transmitted message.

Transmitter

The transmitter shown in Figure 2 will generate the signal that will be transmitted through the channel. Demultiplexing, quadrature amplitude modulation (QAM) and orthogonal frequency division multiplexing (OFDM), and up mixing will work together to create the transmitter signal output.

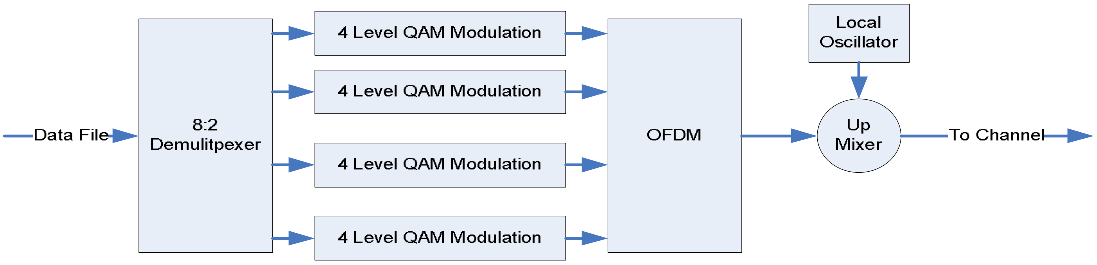

Figure 3 – Transmitter Subsystem Detailed Diagram

Demultiplexing & Modulation

The demultiplexing block will take 8 bits of binary data and then break the 8-bit stream into four 2-bit streams. These 2-bit streams will each be fed into a QAM modulation channel. Once the QAM channels have modulated the input data, they will be passed into the OFDM block. The OFDM system will multiplex the QAM signals together to produce the final modulated output.

Up Mixer

Mixing is done to meet the bandwidth requirements of the channel. The up mixer will increase the frequency of the OFDM signal by multiplying it by a greater carrier frequency. The OFDM signal will be imbedded in the carrier signal that the local oscillator produces. The output of the mixer will be in bandwidth of the channel.

Channel

Figure 3 shows the detail of the channel block from figure 1. The channel block will implement a model of an actual transmission channel. Different parts of the channel will model different channel effects on the output of the transmitter. These are channel gain, multi-path interference, and noise.

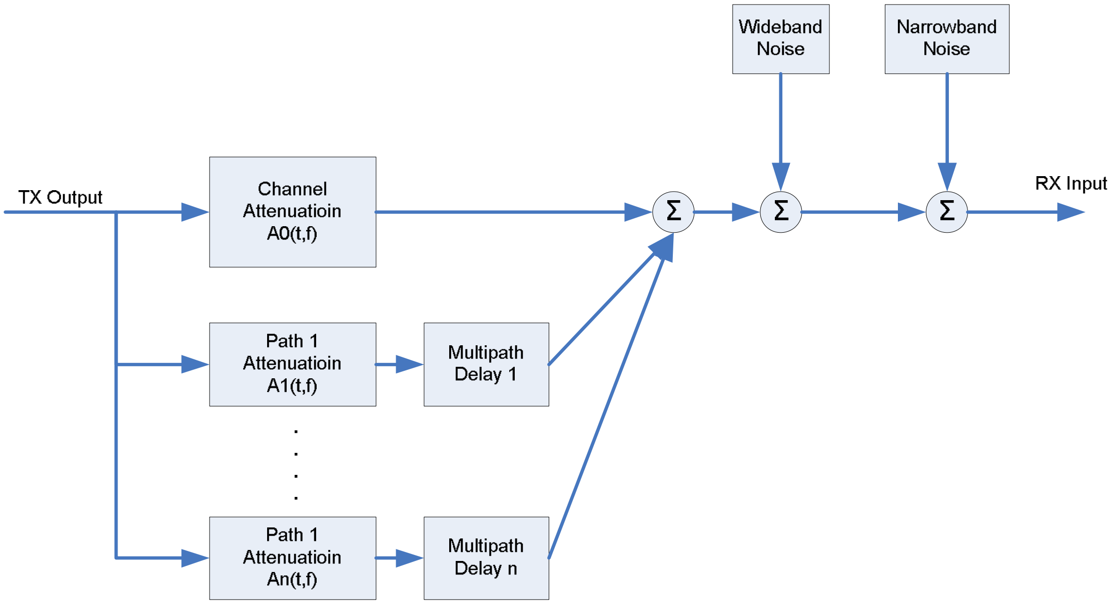

Figure 4 - Detail of Channel

Channel Attenuation

The channel attenuation will model the attenuation or possibly the gain effect that the channel will have on the transmitted signal. It this gain can vary with frequency and/or time.

Multi-path interference

The multi-path interference will model reflections of the transmitted signal. These reflections will arrive at the receiver at different times. Each one of these paths can have its own attenuation that can vary with time and frequency.

Noise

Noise will also be introduced into the signal. This noise is either specific to a limited frequency range (narrow band noise) or affects the whole spectrum of the transmitted signal.

Receiver

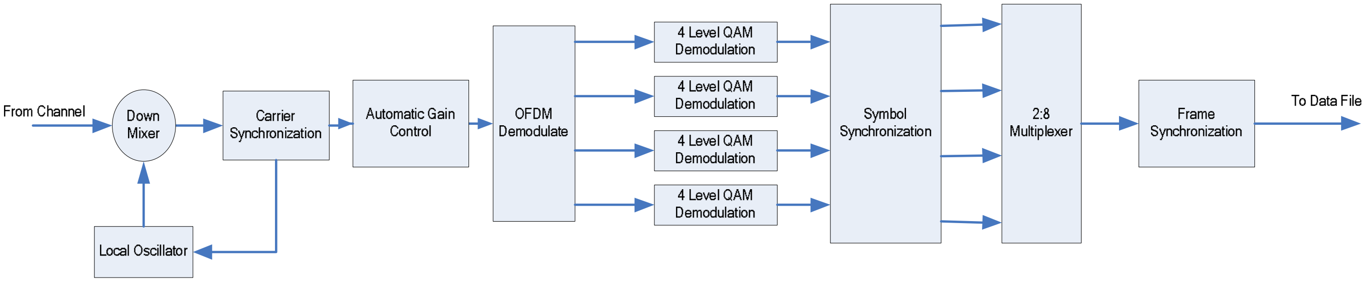

Figure 5 - Receiver Subsystem Detail

The receiver subsystem shown in figure 4 will recover the sent message. To do this it needs to extract the carrier, symbol, and frame timing from the signal. It then will use this information to extract the message from the phase, frequency, and amplitude noise of the channel. The receiver has the following main parts: carrier synchronization, automatic gain control, demodulation, symbol synchronization, and frame synchronization.

Carrier synchronization

The carrier synchronization subsystem will correct for frequency differences between the transmitter and the receiver. It will also correct for the phase delay introduced by the channel. It will do this using a phased locked loop (PLL) technique.

Automatic Gain Control (possibly not implemented)

The automatic gain control system will correct for the time varying differences in channel attenuation. It will do this by adjusting the average power of the input signal to a known value.

Demodulation

The demodulation system consists of two parts: OFDM demodulation and QAM demodulation. The OFDM demodulation will demodulate the signal into its constituent QAM sub signals. The QAM demodulation will demodulate the QAM carriers back into the pulses that were used to modulate it.

Symbol Synchronization

The symbol synchronization will figure out the most appropriate time to sample the pulses coming from the QAM modulation. This will allow the most accurate information to be extracted from the pulse stream. The output of this block will then be fed into a multiplexer. The output of the multiplexer will be the digital data that was fed into the system to begin with.

Frame Synchronization

The frame synchronization will synchronize the data frames of the system. This will align the start time of the message so that the digital data can be interpreted correctly. This will allow the compute file or message to be translated back into its original form.

Preliminary Work and Results

Preliminary work on the software radio project has begun over the course of the fall semester. This has consisted of standards and patent research, getting familiar with the TI board and its Simulink interface, QAM transceiver design, and OFDM transceiver design.

Standards and Patents Research

Standards and patent research was completed to gather information and ideas that would help in the design and implementation of the project.

Standards

The primary standard that this project is based on is IEEE 802.11a. IEEE 802.11a is a standard for wireless networking. In the standard there are details for the implementation and operation of a digital radio for use as a network interface. Our project is an adapted version of this standard, scaled to the scope of the project and the time available for the project.

Patents

The following patents and patent applications were discovered when researching patents and patent applications applicable to this project.

Patent Number	Description
6,091,765	Reconfigurable radio system architecture
6,353,640	Reconfigurable radio frequency communication system
6,937,877	Wireless communication with a mobile asset employing dynamic configuration of a software defined radio
6,954,628	Radio receiver
Patent Application	
2005243952	Methods for processing a received signal in a software defined radio (SDR) system, a transceiver for an SDR system and a receiver for an SDR system
20050190827	Modulation/demodulation apparatus for the encoding and decoding of data and method for encoding and decoding data

Table 1 - Applicable patents and patent applications

Datasheet

Parameter	Symbol	Limits			Units
		Min	Typical	Max	
Modulation Frequency	Fmod		10K Hz		Hz
Data Rate	R		1K Hz		bits / sec
Sampling Rate	Fsamp		100K Hz		Hz

Table 2 - Specifications for Software Radio

Table 2 shows the specifications decided upon for the software radio. The sampling rate of 100KHz is consistent with the sampling frequency of the A/D converter shown in table 3.

TI 6713DSK Board and Simulink

The first half of lab time this semester was used to become familiar with using Simulink to program the TI 6713DSK board. When everything is setup properly with the correct versions of the software, the programming of the board works seamlessly. Code Composer Studio, the programming software that comes with the DSK board, must be running before compilation of the program is to take place. After the target dialog is properly configured (noted in notebook), all one must do is press the compile button. After some time of code generation and compilation, the program is running on the DSK board.

QAM

Quadrature amplitude modulation is a modulation scheme that creates a modulation signal from a binary bit stream. The binary data is broken up, and then composed into two signals which are called quadrature and in-phase signals. The in-phase signal represents the first half of the binary segment, and the quadrature signal represents the other half. The two signals are then mixed with a local oscillator carrier frequency and then summed together. The output from the summer is the QAM output which can be written as:

$s(t) = x(t)\cos(w_c t) - y(t)\sin(w_c t)$

Equation 1 - Equation defining a QAM signal

x(t) is the in-phase component and y(t) is the quadrature component. This can be rewritten in complex envelope form as:

$g(t) = x(t) + jy(t)$

Equation 2 - Complex envelope of QAM signal

A QAM constellation can be designed to identify what symbols represent certain binary values. A simple constellation is shown in Figure 6.

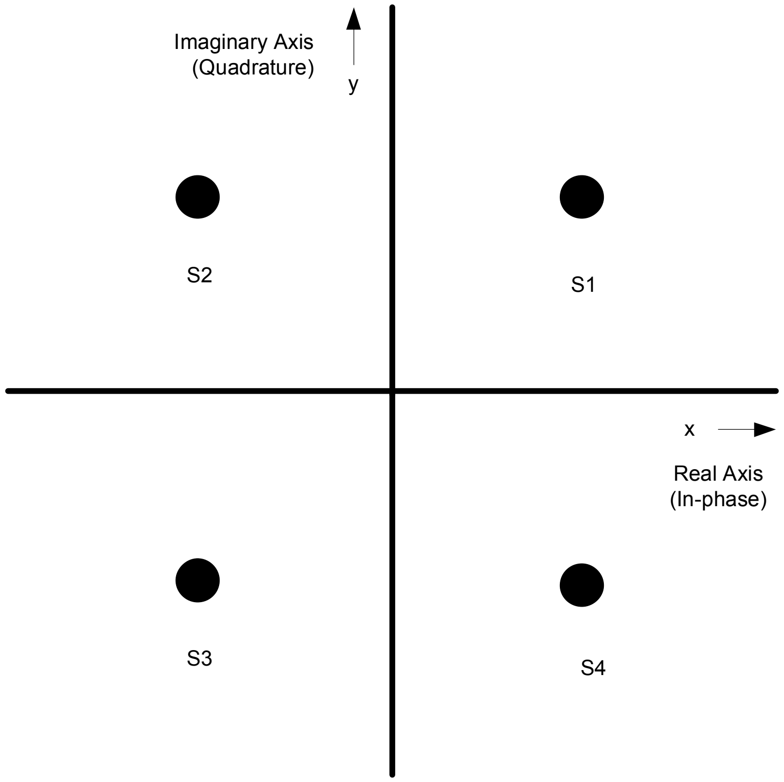

Figure 6: 4 Level QAM Constellation

Preliminary Design

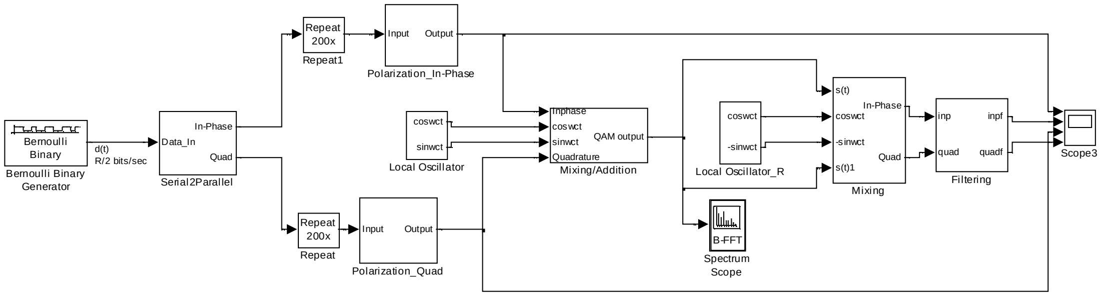

Figure 7 - QAM Transmitter & Receiver

Figure 7 shows the overall block diagram for the QAM transmitter and receiver. It was made and simulated in Simulink.

Results

Currently the local oscillators are operating at 1KHz which can be seen in Figure 8 and the with a bandwidth of 200Hz. The bit rate is set currently at 1/100 sec.

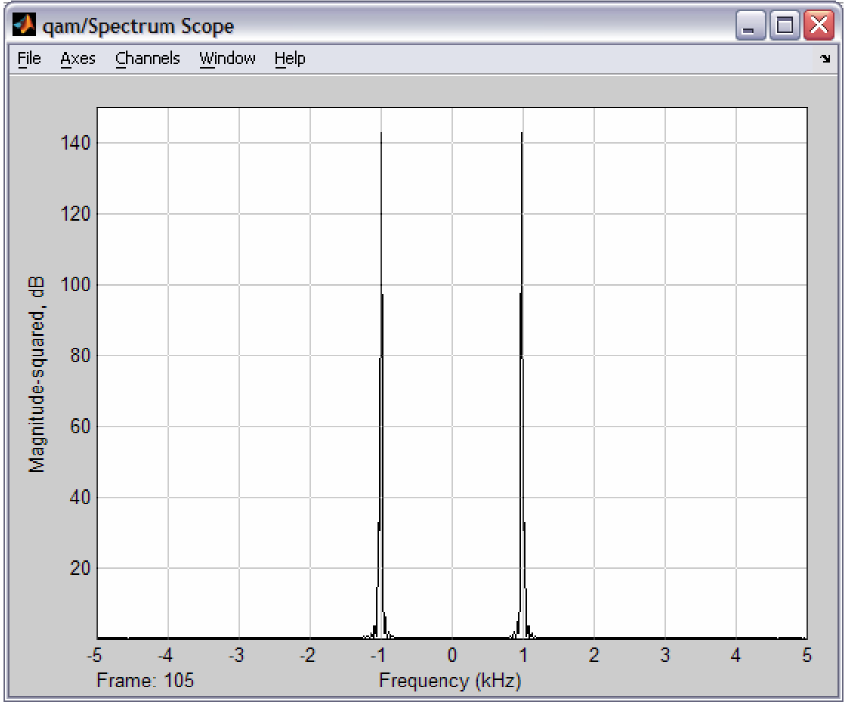

Figure 8 - FFT of QAM transmission signal

The results from the QAM transmitter and receiver match up very well as shown in Figure 9. A digital synthesizer still needs to be implemented on the receiver side. Expansion upon the basic 4-level QAM scheme will also be looked into.

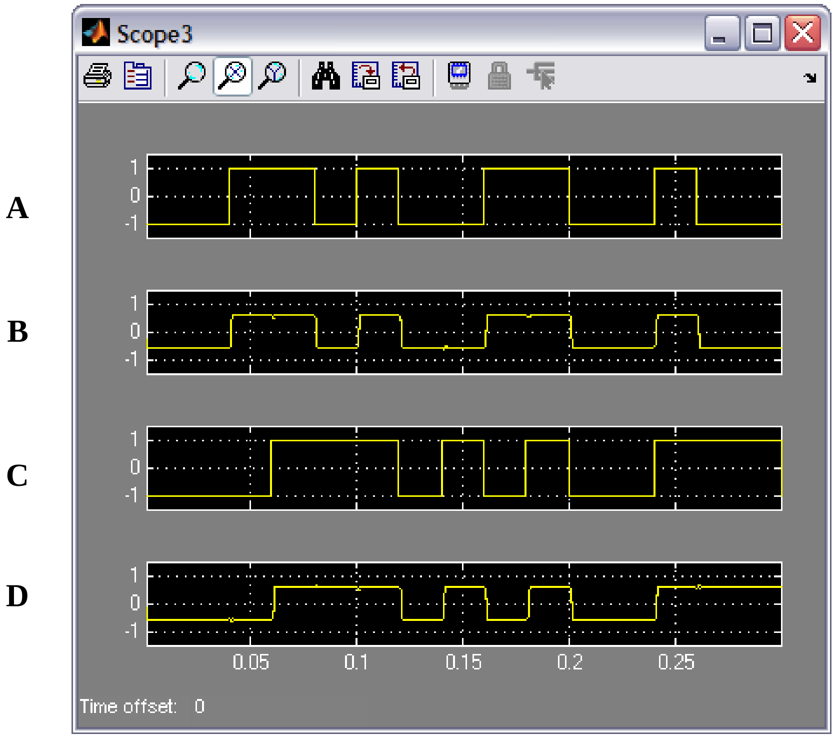

Figure 9 - A. In-phase signal transmitted, B. In-phase output, C. Quadrature signal transmitted, D. Quadrature output

OFDM

Orthogonal Frequency Division Multiplexing or OFDM is a way to transmit many modulated signals at once by multiplexing them over a large number of frequencies. OFDM is different from normal frequency multiplexing because the individual frequency carriers are orthogonal to each other. This allows them to be closely space and not interfere with each other. Leon Couch in his book Digital and Analog Communication Systems gives the complex envelop for OFDM in equation 1.

$$f_n = \frac{1}{T}\left(n - \frac{N-1}{2}\right) \text{ Orthogonal Carrier Frequency}$$

$$\varphi_n(t) = e^{j2\pi f_n t} \text{ - Orthogonal Carrier Function}$$

$$g(t) = A_c \sum_{n=0}^{N-1} w_n \varphi_n(t) \text{ where } 0 > t > T, \ \mathrm{w_n} \text{ is}$$

$$\text{the nth element of the data vector } [w_0, w_1, \ldots w_{N-1}]$$

Equation 3 - OFDM Complex Envelope definition

T is the duration of data symbol on each carrier. According to equation 1, this will create subcarriers 1/T Hz apart. If the part in the parenthesis of the equation for fn is ignored, and the equations are combined the result is the definition for the Inverse Fast Fourier Transform or IFFT. The ignored part is just a frequency shift and the results of this can be accomplished by properly placing the data in the FFT input vector.

Preliminary Design

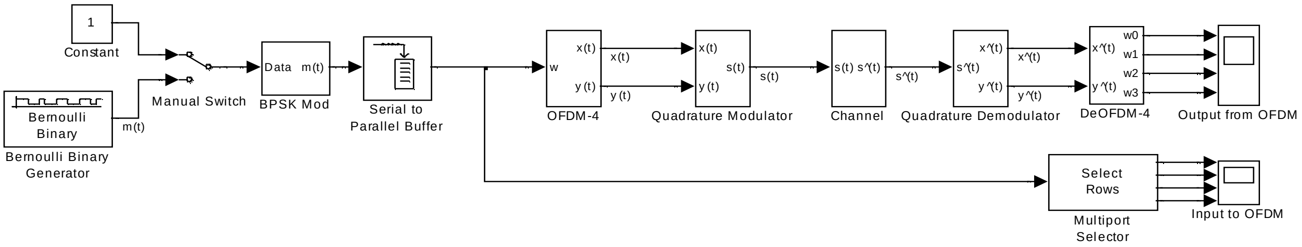

Figure 10 - Overall diagram for OFDM modulator

Figure 10 shows the overall structure for the OFDM modulator. Either a random data source or a constant data source was used as test data. To test the OFDM itself, each carrier had to be modulated also. A simple BPSK (Binary Phase Shift Keying) arrangement was used to test the OFDM modulator. The "BPSK Mod" in figure 10 generates the complex envelope for the BPSK signal from the data source.

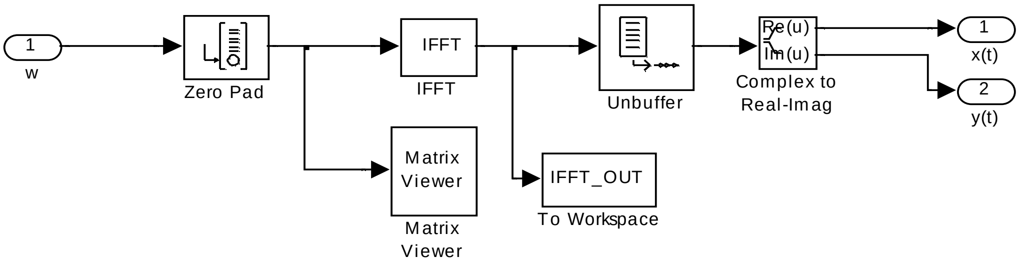

Figure 11 - Block Diagram for the OFDM modulator

Figure 11 shows the block diagram for the transmitter. In figure 10 the input signal was buffered to make the input to the OFDM modulation block four parallel symbols per symbol period of 10mS. Then in the OFDM modulation block the input was zero padded so that the signal will be at the sampling rate of 800 Hz when unbuffered later. After being zero padded, the signal was sent through the IFFT to create the complex envelope for the OFDM signal. After unbuffering, the signal was separated into inphase and quadrature parts and sent out of the block. The two signals were then sent to a standard quadrature transmitter. This resulting signal was sent through a channel bock (that does nothing) and then sent into the demodulating block which mixes with another 200Hz sinusoidal and separates the signal into inphase and quadrature parts. The resulting inphase and quadrature parts were then sent into the OFDM demodulating block shown in figure 12.

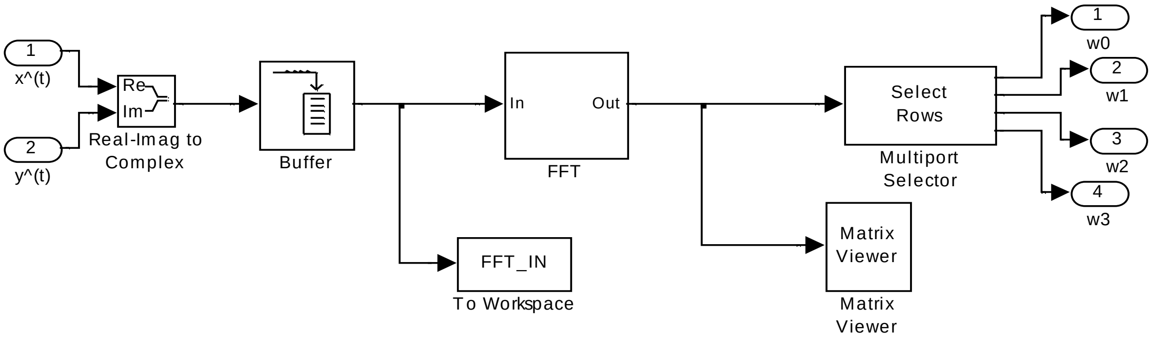

Figure 12 - Block Diagram showing the OFDM demodulator

The demodulation block first combines the inphase and quadrature parts into a complex signal. The signal is then run through a Fast Fourier Transform (FFT) to recover the data.

Results

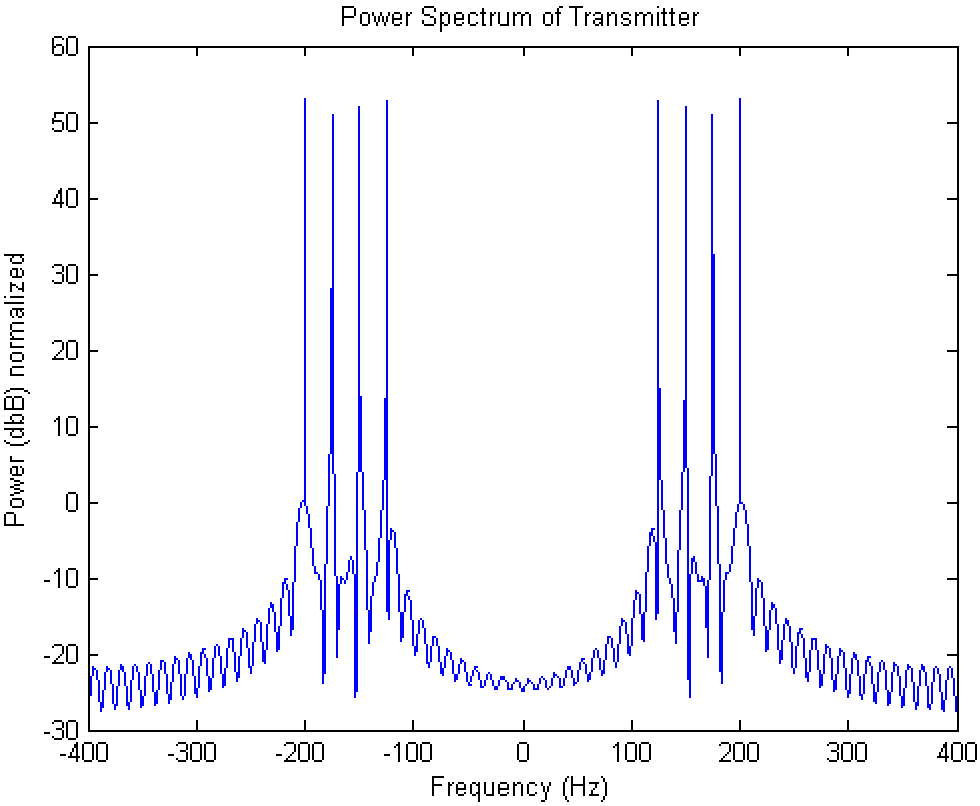

Figure 13 - Resulting signal spectrum when using constant data

Figure 13 shows the resulting signal spectrum from the OFDM transmitter when using constant input data. The result was four carriers spaced by 25 Hz, ending at 200 Hz. The 200 Hz ending frequency is from the modulation frequency of 200 Hz. The reason that the spikes descend in frequency when the data was put in the IFFT in increasing frequency is unknown and needs to be investigated. The frequency spacing is consistent with the symbol rate at the input of the IFFT. The data rate of 100 symbols/second was buffered by 4 which makes it 25 symbols per second. The carrier spacing in OFDM is and should be the symbol rate.

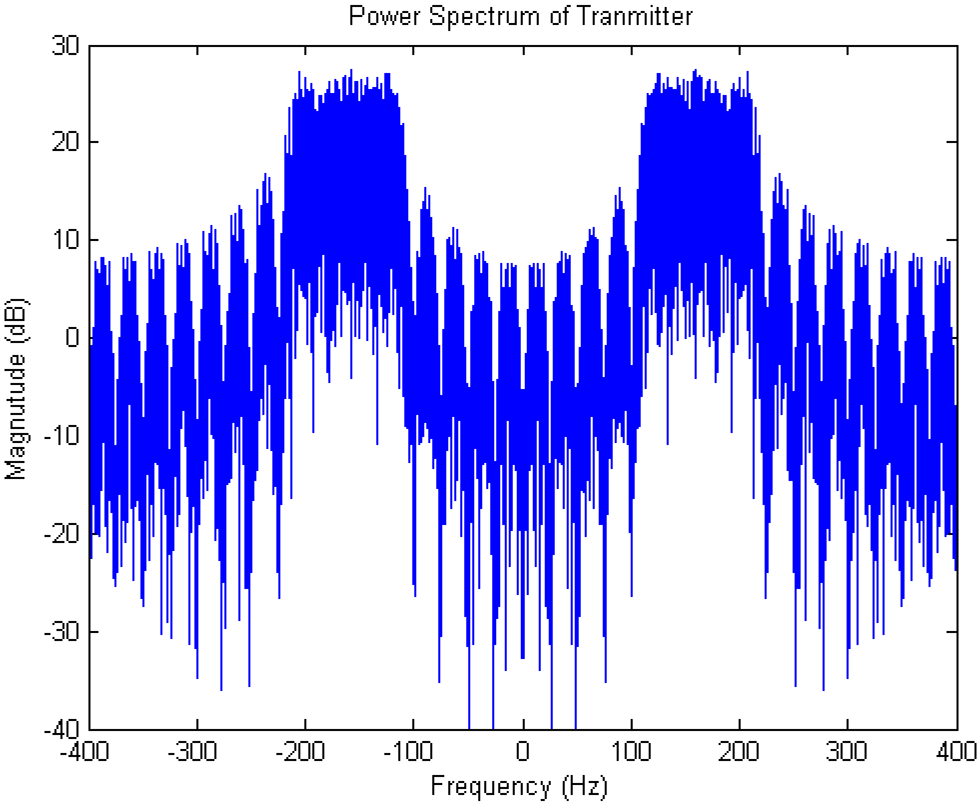

Figure 14 - Power spectrum of OFDM signal using random data

Figure 14 shows the power spectrum of the OFDM signal using random data. This matches closely with the picture found on page 369 of Digital and Analog Communication Systems [1]. Figures 15 and 16 show the parallel input data and parallel output data to the OFDM modulator and demodulator. Although delayed, the amplitude, spacing, shape, and content of the input and output is the same. This signifies a successful demodulation.

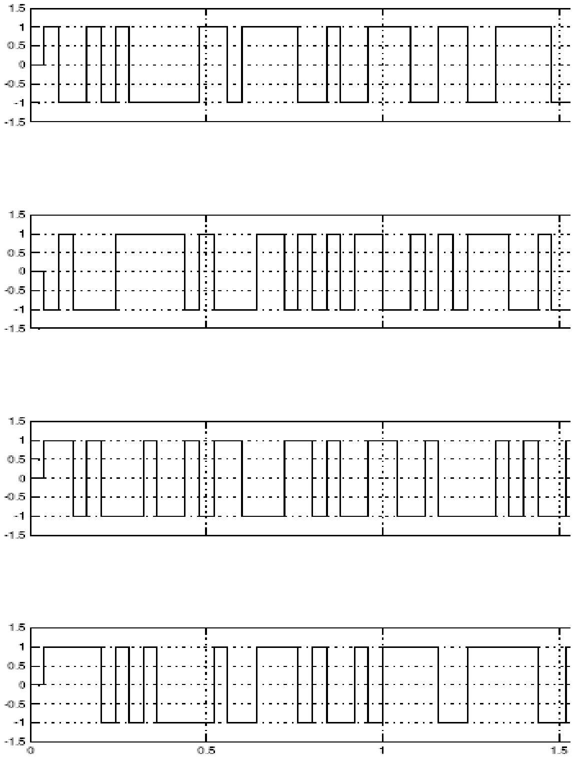

Figure 15 - The 4 input channels to the OFDM tranmitter

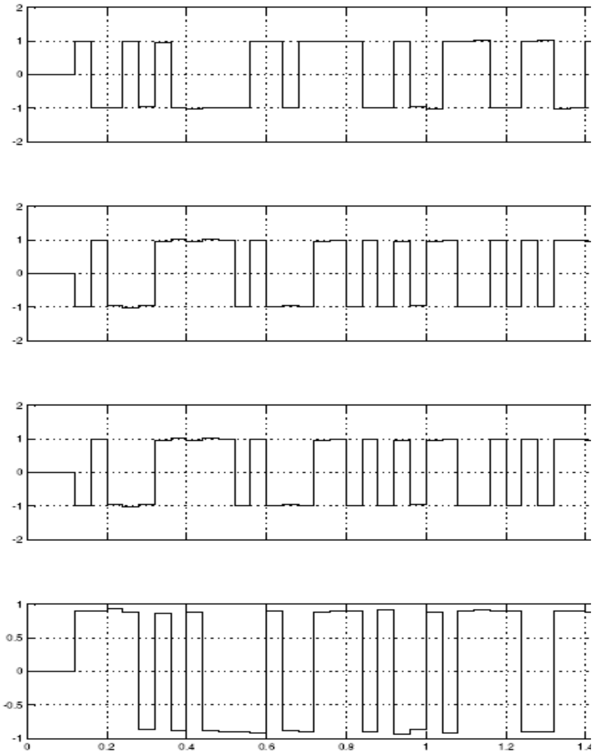

Figure 16 - The 4 output channels from the OFDM receiver

Schedule

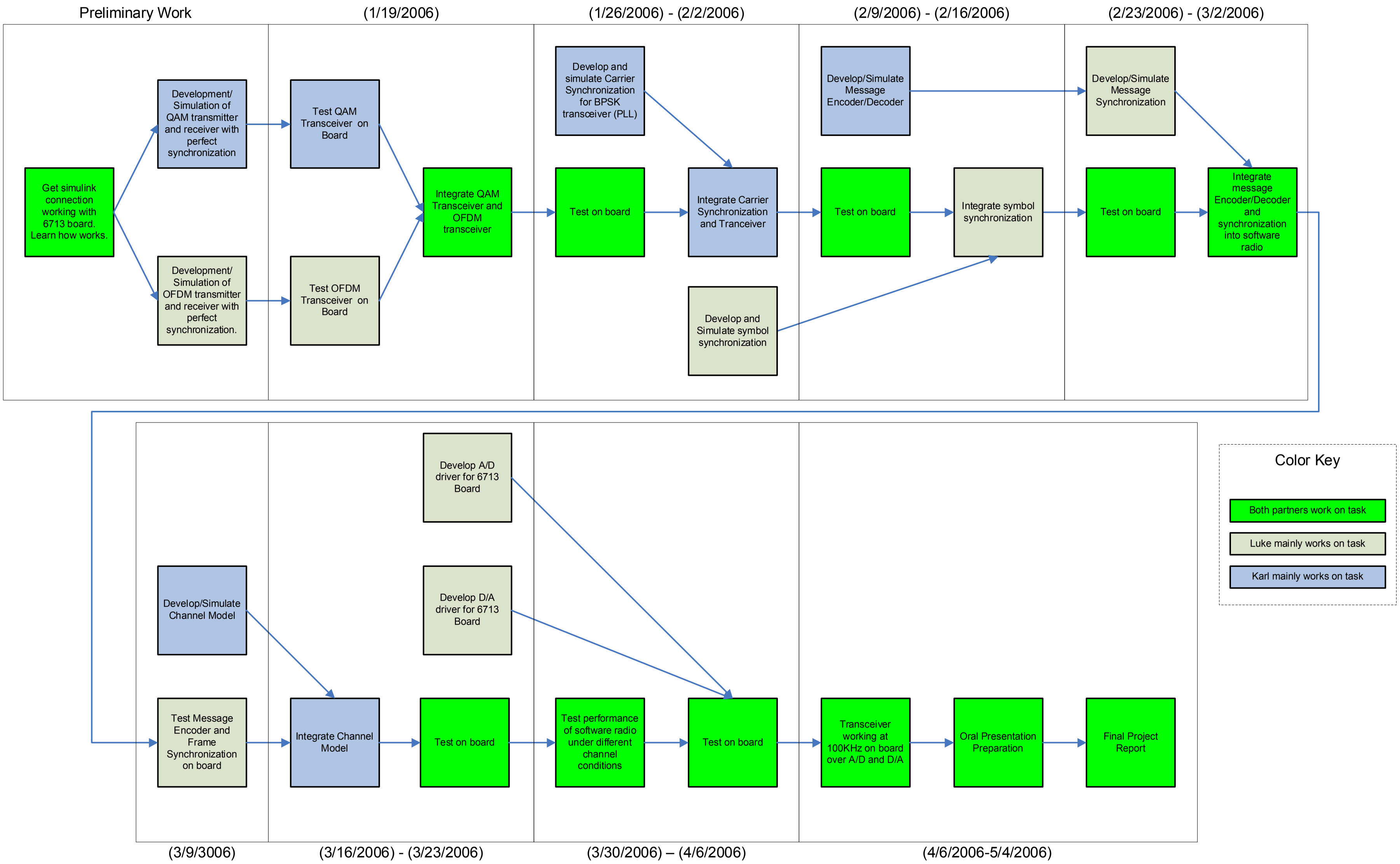

Figure 17 – Project flow and milestones for completion

Figure 17 details the project flow and milestones for completion. The preliminary work has been completed as of writing this document. A date and/or date range is shown above each block. The estimated completion for each

Equipment List

The equipment in table 1 will be needed to complete this project. The equipment has already been ordered and received by the time of writing.

Code	Model	Description	Price ($)
5-6KINTERFACE	DAP9527U	5-6K Interface Board	49
TMDSDSK6713	TMDSDSK6713 (DSP9959U)	DSP Starter Kit	395
TLC4541EVM	TLC4541 (DAP9548U)	16-bit 200KSPS ADC Serial Out	49
DAC8811EVM	DAC8811 (DAP12113U)	16-bit serial input, multiplying D/A converter with single supply 2.7-5.5V. 0.5uS settling time	49

Table 3 - Equipment List

Bibliography

[1] Couch, Leon W. Digital and Analog Communication Systems. 1997. New Jersey: Prentice Hall, 2002.
[3] IEEE Std. 802.11a. “Part 11: Wireless LAN Media Access Control and Physical Layer Specifications.” IEEE SA-Standards Board. June 12, 2003.
[2] Johnson, C. Richard, Jr., Sethares, William A. Telecommunication Breakdown. 2004. New Jersey:Prentice Hall, 2004.